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(56) Documents cited
EP 0041310 A US 4376916 A

(58) Field of search
UK CL (Edition L) H3G GPF GPXX
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(54) Audio signal processing system usable, for example, in a frequency responsive compressor

(57) A plurality of band pass filters are provided for dividing an audio input signal into different pass bands, and level detectors are provided corresponding to the band pass filters for detecting levels of the pass bands passing the respective filters. Each of the level detectors has an individual time constant which is set to be long for a low frequency band and short for a high frequency band. A coefficient is produced dependent on a sum of the outputs of the level detectors. The input signal is multiplied by the coefficient. Use in a compressor is described.

FIG.1

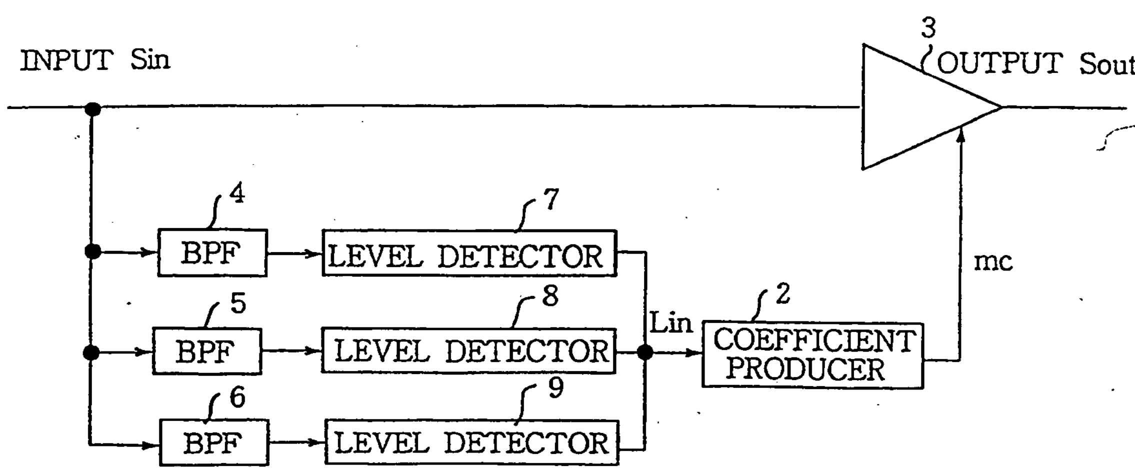

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FIG.1

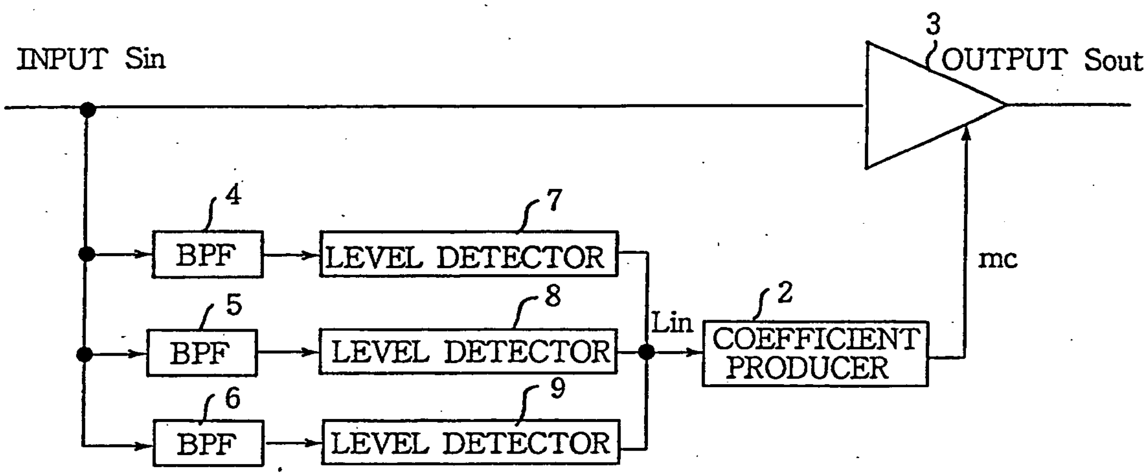

INPUT Sin
3
OUTPUT Sout
4
BPF
7
LEVEL DETECTOR
5
BPF
8
LEVEL DETECTOR
6
BPF
9
LEVEL DETECTOR
Lin
2
COEFFICIENT PRODUCER
mc

FIG.2

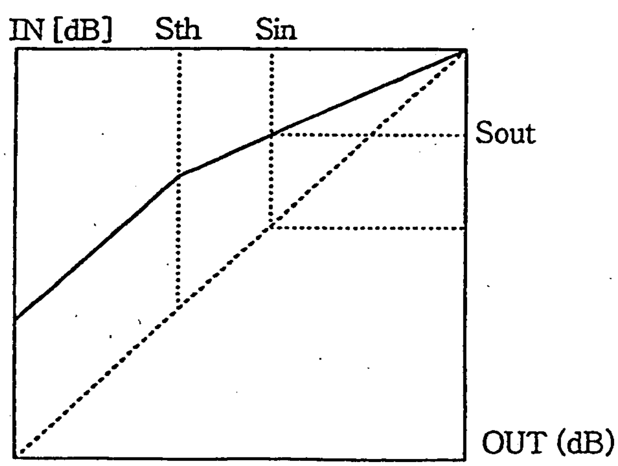

IN [dB]
Sth
Sin
Sout
OUT (dB)

FIG.3

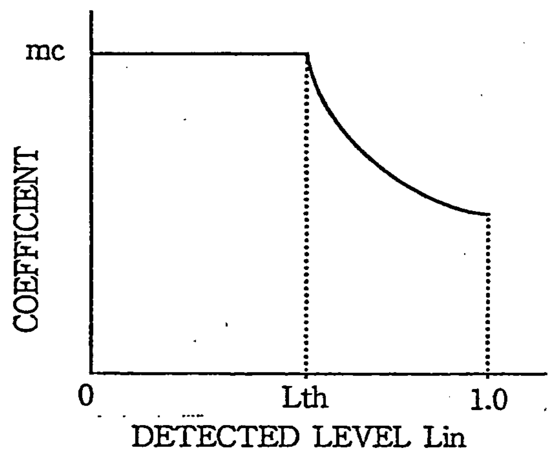

mc
COEFFICIENT
0
Lth
1.0
DETECTED LEVEL Lin

FIG.4

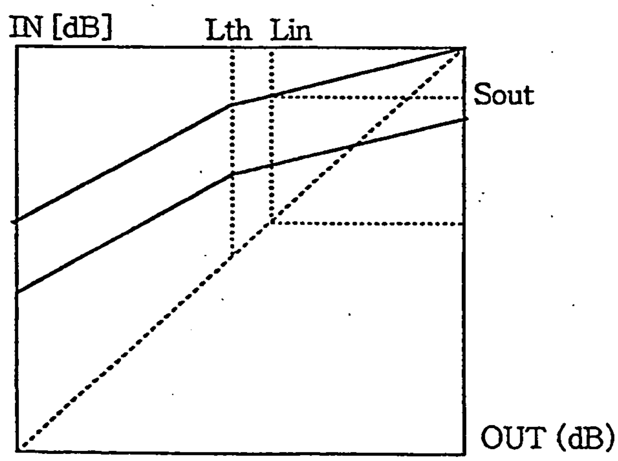

IN [dB]
Lth
Lin
Sout
OUT (dB)

FIG.5

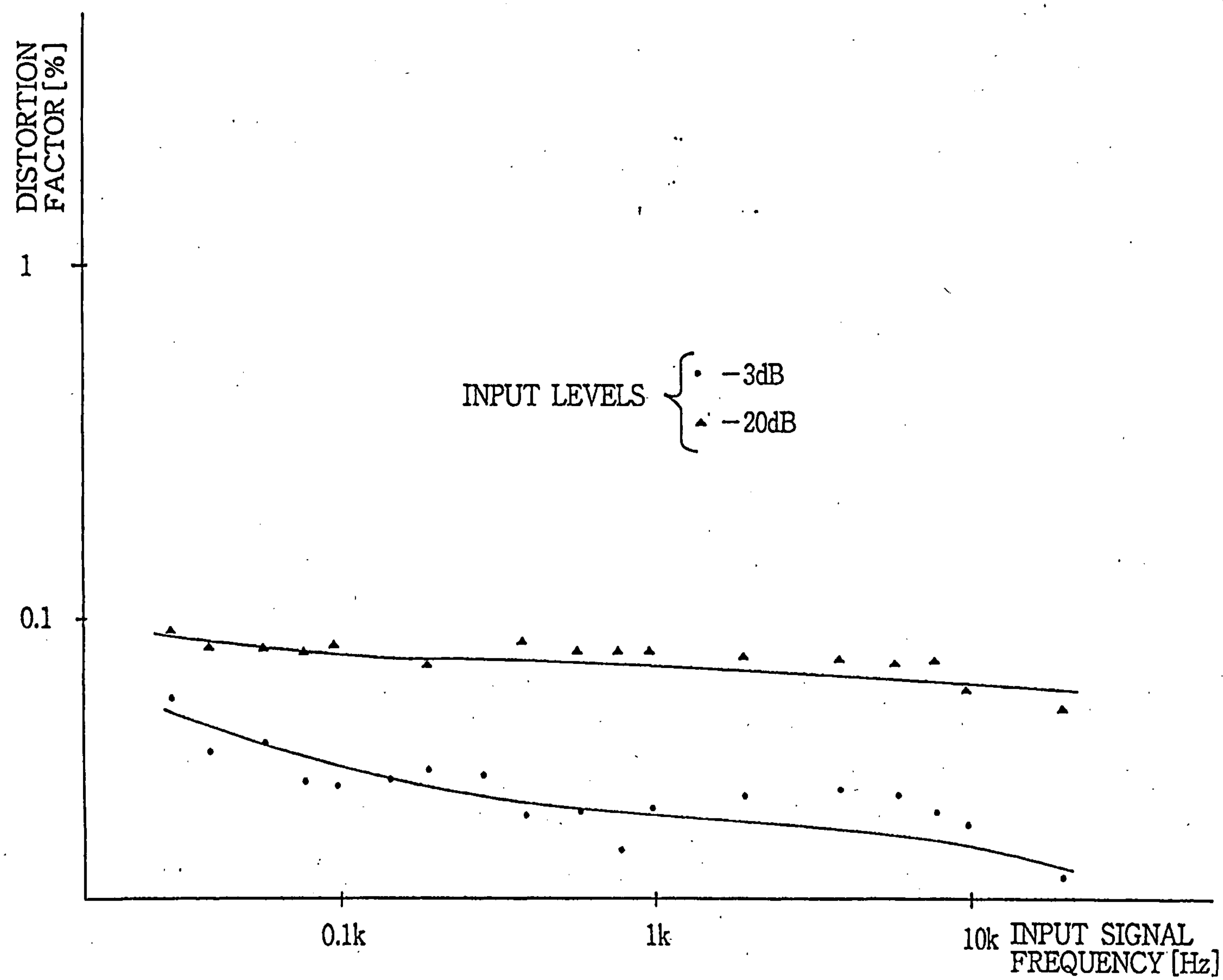
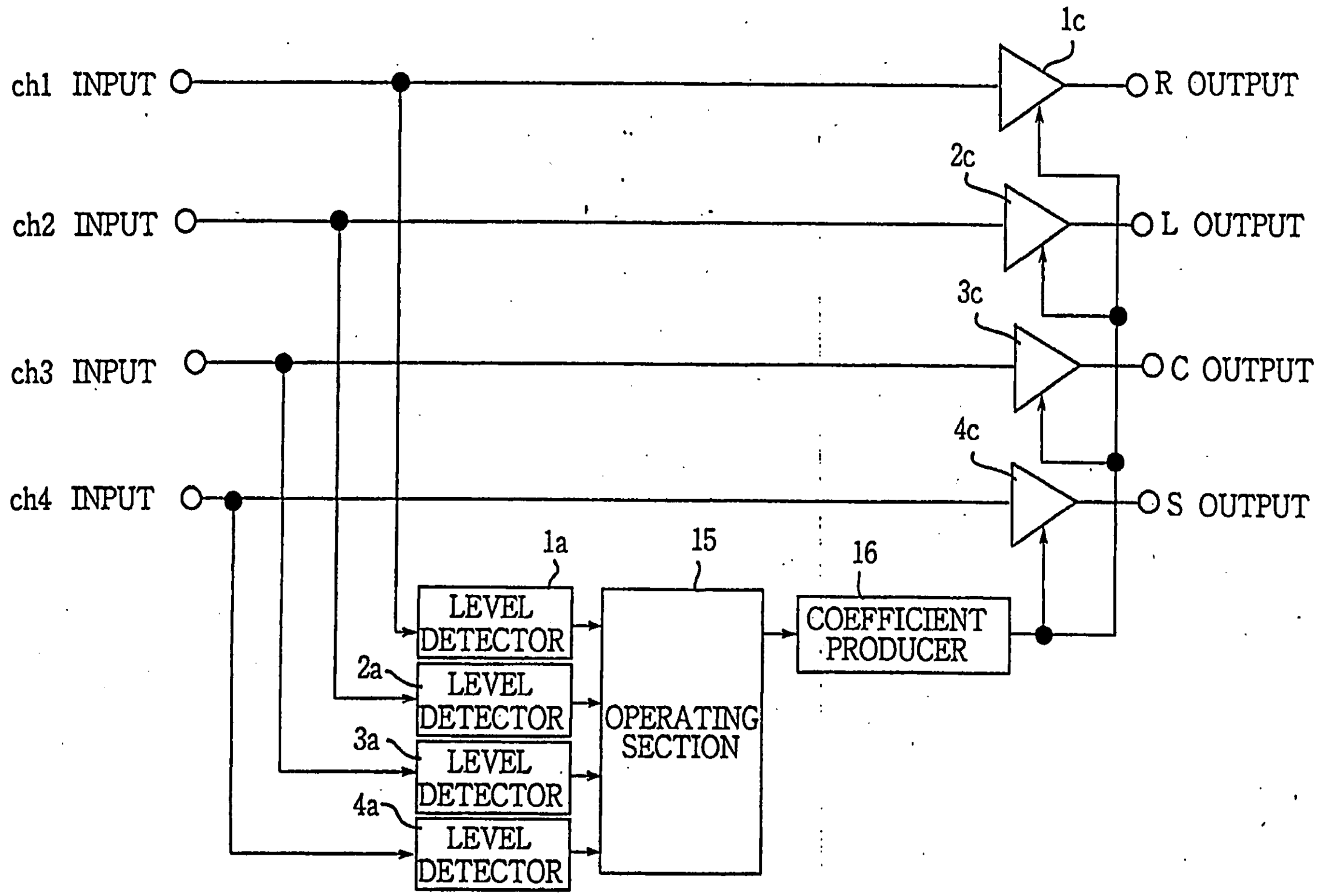

DISTORTION FACTOR [%]
1
0.1
INPUT LEVELS
• −3dB
▲ −20dB
0.1k
1k
10k
INPUT SIGNAL FREQUENCY [Hz]

FIG.6



ch1 INPUT
ch2 INPUT
ch3 INPUT
ch4 INPUT
1c
2c
3c
4c
R OUTPUT
L OUTPUT
C OUTPUT
S OUTPUT
1a
2a
3a
4a
LEVEL DETECTOR
LEVEL DETECTOR
LEVEL DETECTOR
LEVEL DETECTOR
15
OPERATING SECTION
16
COEFFICIENT PRODUCER

FIG.7

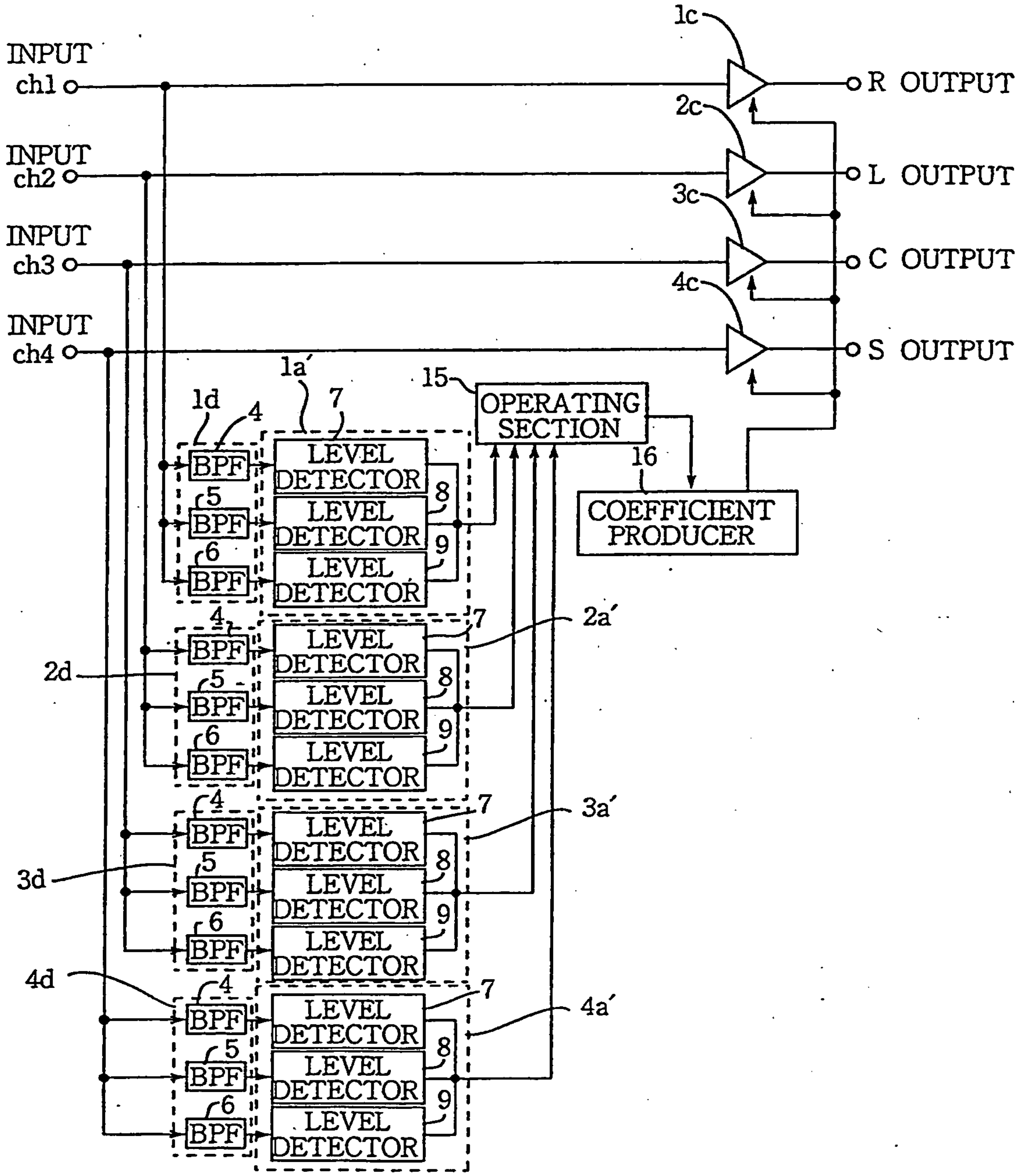
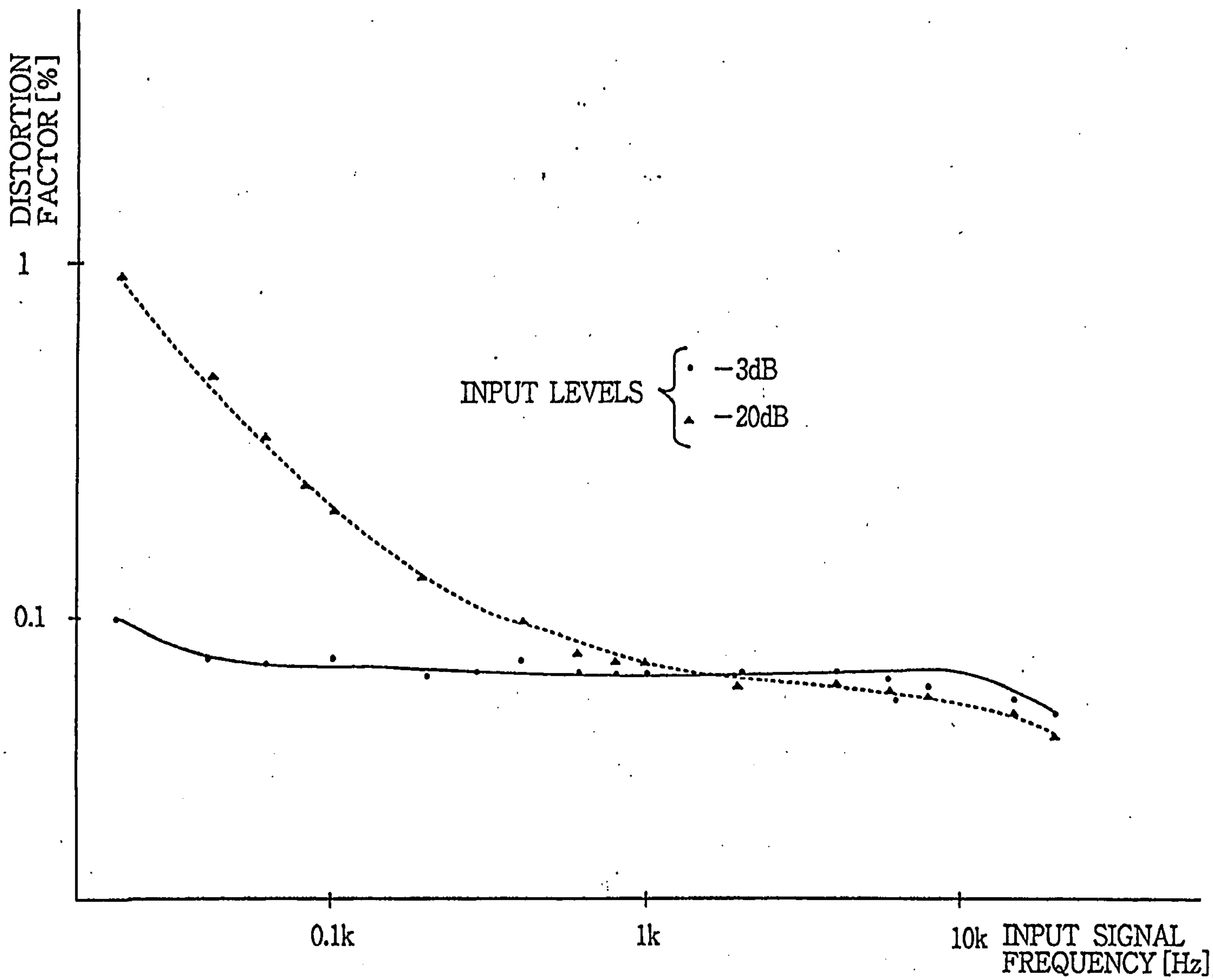

INPUT ch1
INPUT ch2
INPUT ch3
INPUT ch4
1c
2c
3c
4c
R OUTPUT
L OUTPUT
C OUTPUT
S OUTPUT
15
OPERATING SECTION
16
COEFFICIENT PRODUCER
1a′
1d
2d
3d
4d
2a′
3a′
4a′
4
5
6
7
8
9
BPF
LEVEL DETECTOR

FIG.8



DISTORTION FACTOR [%]
1
0.1
INPUT LEVELS
−3dB
−20dB
0.1k
1k
10k
INPUT SIGNAL FREQUENCY [Hz]

6/9

FIG.9

PRIOR ART

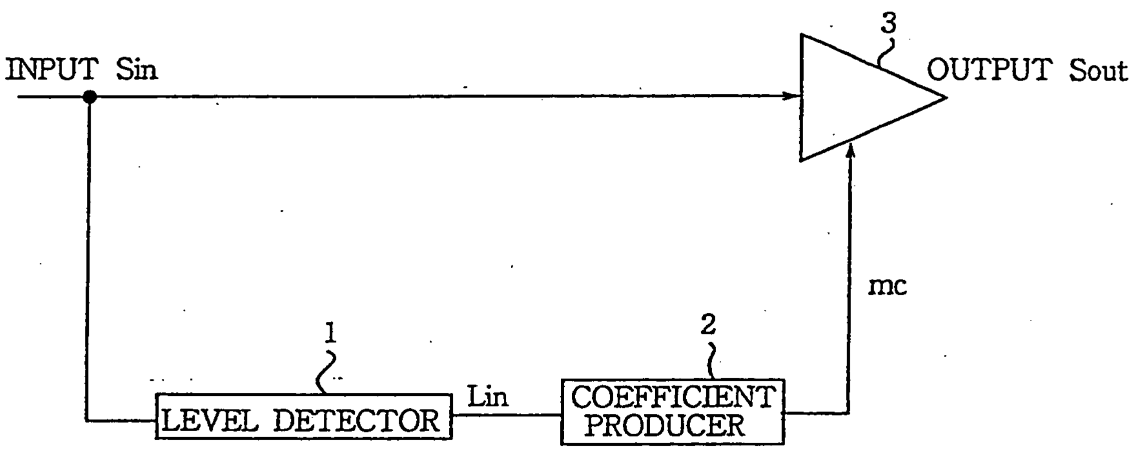


FIG.10 a

PRIOR ART

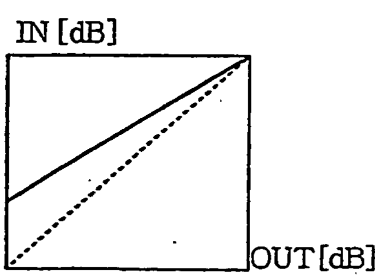


FIG.10 b

PRIOR ART

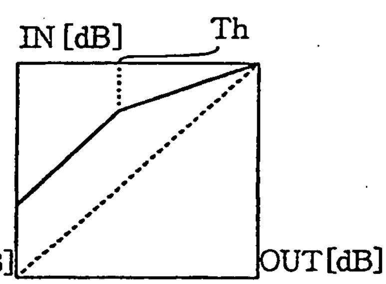


FIG.10 c

PRIOR ART

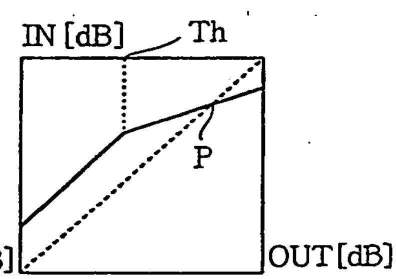


FIG.11

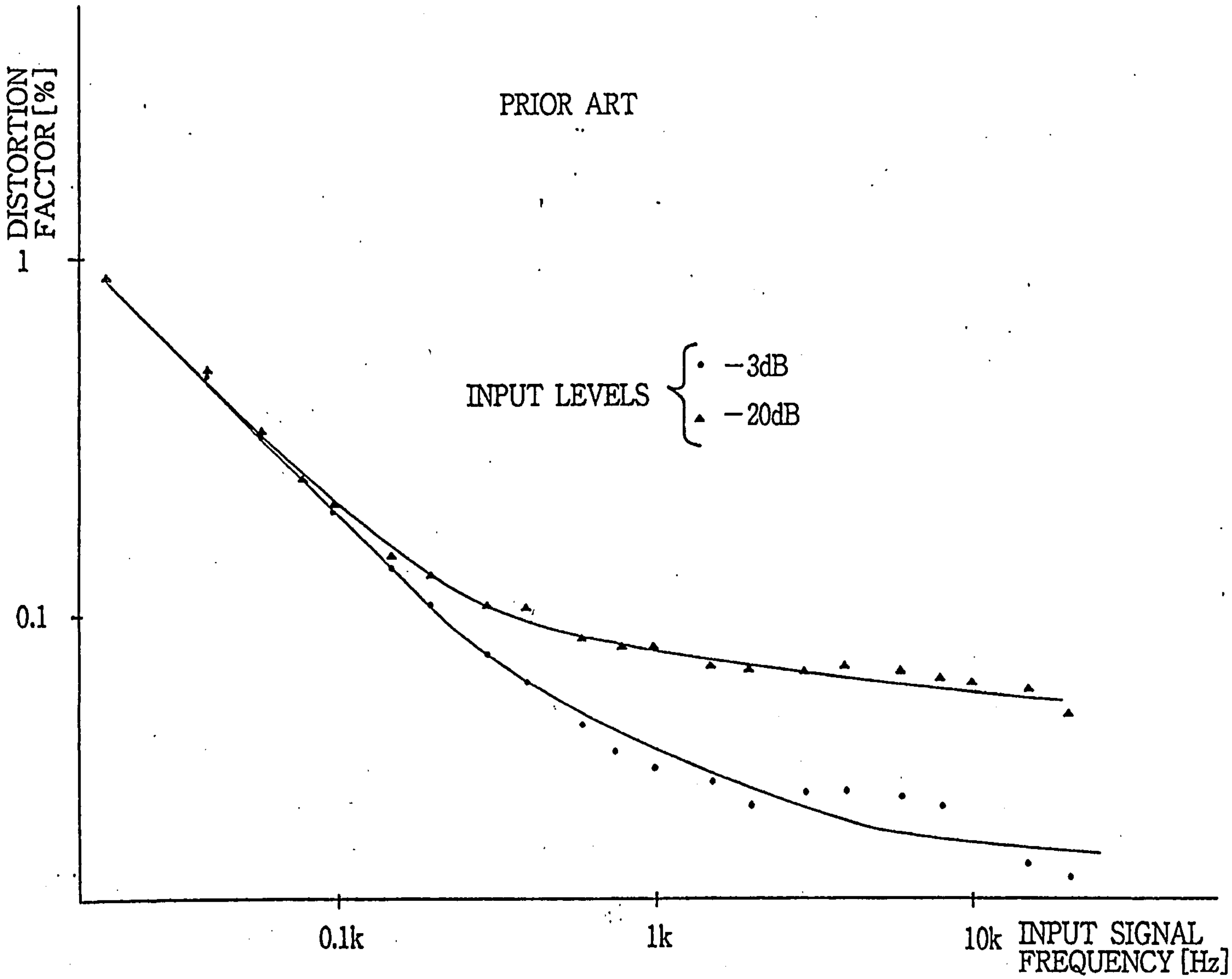
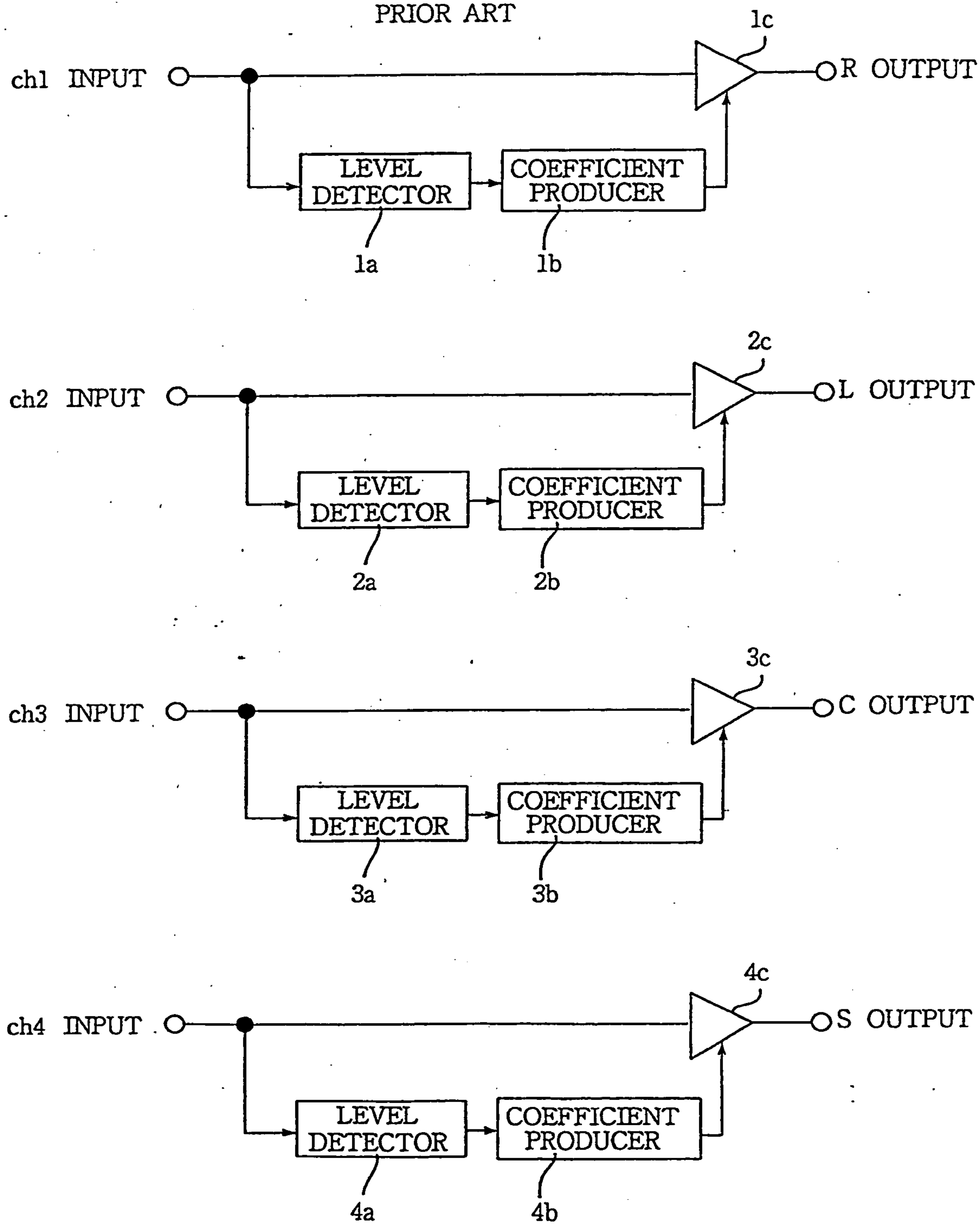

PRIOR ART
DISTORTION FACTOR [%]
1
0.1
INPUT LEVELS
• −3dB
▲ −20dB
0.1k
1k
10k
INPUT SIGNAL FREQUENCY [Hz]

FIG.12

PRIOR ART



AUDIO SIGNAL PROCESSING SYSTEM

The present invention relates to a system for processing an input audio signal, and more particularly to a system for controlling a level of the input audio signal.

An audio signal processing system is provided with a level control device for controlling the level of the input audio signal. There are various types of level control devices such as a compressor, noise gate, limiter and expandor.

Fig. 9 shows a basic construction of a compressor. The compressor has a level detector 1 applied with an audio input signal Sin for detecting the level of the input signal and for producing an output voltage Lin depended on the level, and a coefficient producing means 2 applied with the detected level Lin for producing a coefficient mc corresponding to the detected level Lin. The coefficient producer 2 applies the coefficient mc to a multiplier 3 where the input signal Sin is multiplied by the coefficient mc and an output signal Sout is produced. Thus, the audio input signal is compressed through the compressor.

Figs. 10a to 10c are diagrams showing input and output characteristics of the compressor for compressing the dynamic range of the output signal. As shown in Fig. 10a, as the level of the input signal becomes small, the coefficient mc becomes large, as shown by the solid line. However, in practice there is a maximum value in coefficient. Therefore, as shown in Fig. 10b, the level of the input signal larger than a threshold level Th is compressed. The input signal smaller than the threshold level is multiplied by a predetermined constant coefficient.

Fig. 10c shows another characteristic where a margin is provided on the maximum line. A point P where the level of the input signal is equal to that of the output signal is lower than a full scale level. Thus, the level of input signal larger than the point P is attenuated and the input level smaller than the point P is amplified.

The level detector 1 is provided with a low pass filter (LPF) having different time constants at the attack time (positive-going time constant) and at the recovery time (negative-going time constant). The time constant at the attack time is set to be fast and that of the recovery time is set to be slow. Namely, the positive-going time constant is short and the negative-going time constant is long.

If the time constant is short, the response becomes good, so that the accuracy for detecting the level is increased. However, the detected level is liable to fluctuate, resulting in deterioration of distortion factor as a whole.

Fig. 11 shows characteristics between the distortion factor and the frequency in the conventional system where the attack time is 2ms, the recovery time is 100ms, a compression ratio is 0.7, a compression margin is 5dB, and the levels of input signals are -3dB and -20dB.

It will be seen that at a low frequency level where the input signal is lower than 1 KHz, the distortion factor of the detected output becomes high.

An object of the present invention is to provide an audio signal processing system which may prevent the detected signal from distorting.

According to the present invention, there is provided an audio signal processing system comprising a plurality of band pass filters for dividing the audio signal into different pass bands, a plurality of level detectors provided corresponding to the band pass filters for detecting the levels of the pass bands passing the respective filters, each of the level detectors having an individual time constant which is

set to be long for a low frequency band and short for a high frequency band, coefficient producing means for summing the outputs of the level detectors and for producing a coefficient in dependency on the summed output, level control means applied with the coefficient for controlling the level of the input signal by the coefficient.

In the present invention, the deterioration of the distortion factor when the level of the input signal is detected is eliminated. The input signal is divided into a plurality of bands with the band pass filters and the divided bands are detected with the level detectors, each having an individual time constant corresponding to the pass band.

The time constant is set to be long for a low frequency band and set to be short for a high frequency band. The maximum time constant is a value as small as possible without deteriorating the distortion factor. Thus, a response of the input signal to the low and high frequency bands are improved.

The other objects and features of this invention will become understood from the following description with reference to the accompanying drawings.

Fig. 1 is a block diagram showing an audio signal processing system according to the present invention;

Fig. 2 shows a diagram for explaining characteristics of an input signal and an output signal;

Fig. 3 is a diagram showing a characteristic of a coefficient in a coefficient controller;

Fig. 4 is a diagram showing input and output characteristics of a compressor;

Fig. 5 shows graphs of frequency characteristics of different levels of the input signal corresponding to a distortion factor;

Fig. 6 is a block diagram showing a second embodiment of the present invention;

Fig. 7 is a block diagram showing a third embodiment;

Fig. 8 shows graphs of frequency characteristics of different levels of the input signals for a conventional system and the system of the present invention corresponding to a distortion factor;

Fig. 9 is a block diagram showing a conventional audio signal processing system;

Figs. 10a to 10c are diagrams showing characteristics of input signal and output signal of the conventional system;

Fig. 11 shows graphs of frequency characteristics of the different levels of the input signal

corresponding to a distortion factor according to the conventional system; and

Fig. 12 is a block diagram showing another conventional system.

Referring to Fig. 1 showing a compressor as an audio signal processing system according to the present invention, the same structures as the conventional system are identified with the same reference numerals as Fig. 9.

The compressor comprises three band pass filters (BPF) 4, 5 and 6 applied with the input signal Sin. Each of the BPFs 4, 5 and 6 has a different pass band which is set as follows.

pass band of the BPF 4 : 200 Hz or less

pass band of the BPF 5 : between 200 Hz and 2 KHz

pass band of the BPF 6 : 2 KHz or more

Three level detectors 7, 8 and 9 are provided corresponding to the BPFs 4, 5 and 6, respectively. In order to detect a waveform, there are three detecting methods such as peak value detection, a means value detection, and an rms value detection. The level detector of the present invention uses the mean value detection and has a low pass filter (LPF) of a low cut off frequency for smoothing the level of the input signal. The attack time and the recovery time of the

LPF have different time constants with dynamic characteristics corresponding to the respective pass bands of the BPFs 4, 5 and 6. These time constants are set as follows.

	attack time	recovery time
level detector 7	40 msec	2 sec
level detector 8	6 msec	300 msec
level detector 9	2 msec	100 msec

The maximum time constant 40 msec at the attack time is a value as small as possible without deteriorating the distortion factor.

If the attack time and the recovery time are set to the same time constant irrespective of the pass band, the smoothing accuracy decreases particularly in a low frequency level. When the level of the signal is detected, a ripple generates in the detected signal, which deteriorates the distortion factor. Consequently, in the level detector of the present invention, the time constants at attack time and recovery time in the low frequency are set to be longer.

The detected levels Lin are applied to the coefficient producing means 2. The coefficient mc for obtaining the output signal Sout is set in accordance with parameters such as the level of the input signal Sin, a compression ratio k, and a threshold level Sth.

Referring to Fig. 2, if the level of the input signal Sin is larger than the threshold level Sth, the dynamic range of the output signal Sout is compressed corresponding to that of the input signal with the compression ratio k ($0 \leqq k \leqq 1$). The output signal Sout is represented as follows.

$$Sout = Sin^{k}$$

Since the output signal Sout is obtained by multiplying the input signal Sin by the coefficient mc, the coefficient mc is obtained by an equation as follows.

$$mc = Sin^{k-1}$$

To the contrary, if the input signal Sin is smaller than the threshold level Sth, the output signal is represented as follows.

$$Sout = cSin \text{ (c is a constant)}$$

Since the constant c is equal to the coefficient mc at the threshold level Sth, the constant c is represented as

$$c = Sth^{k-1}$$

From the foregoing, the output signal Sout is obtained by equations as follows.

$$mc = Sin^{k-1} \ (Sin \geqq Sth)$$
$$mc = Sth^{k-1} \ (Sin \leqq Sth)$$

$$Sout = mcSin$$

Furthermore, the coefficient mc has a characteristic as shown in Fig. 3. Namely, the detected level Lin of the level detectors is determined in the range between 0 and 1. If the detected level Lin is smaller than a threshold level Lth (Lin≦Lth), the coefficient mc is constant. If Lin≧Lth, the coefficient mc rapidly reduces as shown by the reducing curve.

Describing the operation of the compressor, the input signal Sin applied to the compressor is divided into three pass bands at the BPFs 4, 5 and 6. The divided bands are detected at the respective level detectors 7, 8 and 9. The detected levels Lin are applied to the coefficient producer 2 where the levels are summed up, and the coefficient mc is controlled in dependency on the total of the detected levels and applied to the multiplier 3. The multiplier 3 multiplies the input signal Sin in accordance with the characteristics shown in Fig. 4. Thus, the output signal Sout is produced by $Sout=mcSin=Lin^{-(1-k)}Sin$.

Fig. 5 shows graphs of the characteristics of the compressor of the present invention where the input signals are the same levels as the conventional system shown in Fig. 11. The levels of the signals are approximately flat and the distortion factor lower than

1 KHz is considerably improved compared with the conventional system.

In the above described embodiment, the number of BPFs may be increased to four or more, thereby more finely dividing the input signal and increasing the accuracy of the coefficient mc.

The present invention may also be applied to the noise gate, limiter and expandor.

In accordance with the present invention, the deterioration of the distortion factor when the level of the input signal is detected is eliminated. The input signal is divided into a plurality of bands with BPFs and the divided bands are detected with the level detectors having different time constants corresponding to the BPFs.

The time constant is set to be long with respect to a low frequency band and set to be short with respect to a high frequency band. The maximum time constant is a value as small as possible without deteriorating the distortion factor. Thus, a response of the input signal to the low and high frequency bands are improved. Since the time constant at attack time and recovery time are set to be longer in the low frequency band, the distortion factor in the low frequency band is improved.

Fig. 12 shows another conventional audio signal processing system for detecting levels of a plurality of input signals fed from a plurality of channels used in a Dolby system. Each of the signals fed from channels ch1, ch2, ch3 and ch4 is applied to a compressor which is the same structure as in Fig. 9. The compressors comprise level detectors 1a, 2a, 3a and 4a, coefficient producing means 1b, 2b, 3b and 4b, and multipliers 1c, 2c, 3c and 4c, respectively. The multiplier 1c produces an output signal R which is applied to a right speaker. The multiplier 2c produces an output signal L which is applied to a left speaker. The multiplier 3c produces an output signal C for a center speaker and the multiplier 4c produces an output signal S for a rear speaker.

However, in the system, since each of the input signals from the four channels ch1 to ch4 is compressed by the individual compressor, the levels of the signals detected at the respective level detectors 1a to 4a are different from each other, and hence the values of the coefficients controlled at the coefficient producers 1b to 4b are also different from each other. Accordingly, the output signals R, L, C and S produced at the multipliers 1c to 4c are different from each other. Thus, sound image reproduced from four speakers differs from the original sound image localization of the

signal source. Such a disadvantage is also eliminated by the system of the present invention.

Referring to Fig. 6 showing the second embodiment of the present invention, a compressor of the second embodiment has an operating section 15 applied with the levels of the input signals detected by the respective level detectors 1a to 4a. The operating section 15 operates to obtain a proper level from the detected levels, for example, a maximum level thereof. The operated maximum value is fed to a coefficient producer 16 where a coefficient is produced in dependency on the maximum value and applied to the respective multipliers 1c to 4c. The multipliers 1c to 4c operate to multiply the input signals from the channels ch1 to ch4 with the same coefficient and produces output signals R, L, C and S. Thus, definite sound image localization is obtained.

In place of the maximum value of the detected level, a minimum value or a mean value can be used.

Fig. 7 shows the third embodiment where the compressor comprises BPF sections 1d, 2d, 3d and 4d to which input signals from the channels ch1 to ch4 are applied. Each of the BPF sections has three BPFs 4, 5 and 6. Lever detector sections 1a′, 2a′, 3a′ and 4a′ are connected to the respective BPF sections 1d to 4d. Each of the level detector sections has three level

detectors 7, 8 and 9. The detected levels of the level detector sections 1a′ to 4a′ are fed to the operating section 15. Other structures are the same as the second embodiment and the same parts thereof are identified with the same reference numerals as Fig. 6. The characteristics of the BPFs 4, 5 and 6 and the level detectors 7, 8 and 9 are the same as those of the first embodiment. The compressor is operated in the same manners as the previous embodiments.

As shown in Fig. 8, the level of the signal of the present invention shown by a solid line is approximately flat compared with that of the conventional system shown by a dotted line, and the distortion factor lower than 1 KHz is considerably improved. Furthermore, quality of the reproduced sound is increased.

While the presently preferred embodiments of the present invention have been shown and described, it is to be understood that these disclosures are for the purpose of illustration and that various changes and modifications may be made without departing from the scope of the invention as set forth in the appended claims.

WHAT IS CLAIMED IS:

1. An audio signal processing system comprising:

a plurality of band pass filters for dividing an audio input signal into different pass bands;

a plurality level detectors provided corresponding to the band pass filters for detecting levels of the pass bands passing the respective filters;

each of the level detectors having an individual time constant which is set to be long for a low frequency band and short for a high frequency band;

coefficient producing means for summing the outputs of the level detectors and for producing a coefficient in dependency on the summed output; and

level control means applied with the coefficient for controlling the level of the input signal by the coefficient.

2. A system according to claim 1 wherein

the level control means is a multiplier for multiplying the input signal by the coefficient.

3. A system according to claim 1 wherein

the input signal comprises a plurality of input signals of different channels, and a set of the band pass filters and a set of the level detectors are provided for each channel so as to produce a coefficient for each channel.

4. An audio signal processing system substantially as described herein with reference to Figs. 1 to 8 of the accompanying drawings.

Patents Act 1977
Examiner's report to the Comptroller under Section 17 (The Search Report)

Application number: GB 9301933.9

Relevant Technical fields

(i) UK Cl (Edition L) H3G (GPF, GPXX)

(ii) Int Cl (Edition)

Databases (see over)

(i) UK Patent Office

(ii) ONLINE DATABASES: WPI

Search Examiner: D MIDGLEY

Date of Search: 29 APRIL 1993

Documents considered relevant following a search in respect of claims

Category (see over)	Identity of document and relevant passages	Relevant to claim(s)
A,&	EP 0041310 A (CBS) Whole document	1
A,&	US 4376916 (CBS) Whole document	1

Category	Identity of document and relevant passages	Relevant to claim(s)

Categories of documents

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E: Patent document published on or after, but with priority date earlier than, the filing date of the present application.

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